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Mobility Impact on the End-to-End Delay Performance for VoIP over LTE

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Abstract

Long Term Evolution (LTE) is the last step towards the 4th generation of cellular networks. This revolution is necessitated by the unceasing increase in demand for high speed connection on LTE networks. This paper focuses on the performance evaluation of End-to-End delay under variable mobility speed for VoIP (Voice over IP) in the LTE network. In the course of E2E performance evaluation, realizing simulation approach three scenarios have been modeled using OPNET 16.0. The first one is the baseline network while among other two, one consists of VoIP traffic solely and the other consists of FTP along with VoIP. E2E delay has been measured for both scenarios in various cases under the varying mobility speed of the node. Simulation results have been studied and presented in terms of comparative performance analysis of the three network scenarios. In light of the result analysis, the performance quality of a VoIP network (with and without the presence of additional network traffic) in LTE has been determined and discussed. The simulation results for baseline VoIP network (non-congested) congested VoIP network and congested VoIP with FTP network show that as the speed of node is gradually increased, E2E delay slightly increases.

Keywords: LTE, VoIP, E2E delay, Throughput and OPNET;

1. Introduction

The trend of the modern society is as the days go by, time is getting more expensive and commodity is getting cheaper. To create a world compatible for this, it is necessary to create a network backbone for the whole world so the information along with communication, is instantaneous. As internet is the main information database, cellular technology is required to merge with the core internet structure, with all its bandwidth and fast trafficking facility in the cheapest way possible. This has been the fundamental premise behind the development of LTE. The eNodeB (e.g. evolved Node B) deployments in LTE (Long Term Evolution) however have several challenges. A numerous eNodeBs and other 3GPP cells can be deployed in macro cellular network. Also the eNodeB is based on self-optimized and self-configured in the network. Therefore, the eNodeB is automatically and frequently turned on/off in the macro-cell coverage with various deployment scenarios. While considering these factors, system should support the

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continuous mobility when the User Equipment (UE) moves to in/out door region of eNodeB. From this viewpoint, mobility of UE in terms of VoIP (Voice over IP) traffic is the key requirement for feasibility of eNodeB deployment in 3GPP LTE system [1]. The study of the performance of VoIP over LTE under varying mobility speed thus has a great significance. Nowadays, communication and network technology have significantly expanded. As LTE is relatively a new technology, there are not enough technical documents to get a deeper knowledge of LTE for real-time application such as voice. Introduction of LTE, the 4th Generation (4G) network technology releases comprising 8 specifications are being finalized by the 3rd Generation Partnership Project (3GPP) [2]. LTE determines peak data rate for Downlink (DL) 100 Mbps and Uplink (UL) data rate for 50Mbps, increased cell edge user throughput, improved spectral efficiency and scalable bandwidth from 1.4 MHz to 20 MHz. VoIP capacity of LTE has to show better performance as Circuit Switch voice of UMTS [3]. LTE should be at least as good as the High Speed Packet Access (HSPA) evolution track also in voice traffic [4]. The core network of LTE is purely packet switched and optimized for packet data transfer, thus speech is also transmitted purely with VoIP protocols. By the same time demand for the higher quality of wireless communications has increased as well. Use of demand driven applications and services have been growing rapidly to satisfy users. Meeting such demand poses a challenge for the researchers to solve till now. Among such demands, enhance quality of voice and data transfer rates are one of the main aspects to improve. Hence, to improve the performance of such important aspects, performance evaluation of VoIP can point out the issues which can be resolved to improve the overall performance of LTE networks. In this paper, VoIP application is used to represent the class of inelastic, real-time interactive applications that is sensitive to End-to-End (E2E) delay but may tolerate packet loss [5]. This need is much more expedient in real-time application such as voice which has enormous importance in providing efficient services in order to fulfill the users expectation, and hence to the researchers to improve the technology to meet the ever growing demand of efficient use of the system. It is expected that LTE should support a significantly higher number of VoIP users. The important factor is now the quality of service (QoS) of VoIP [6]. To measure QoS of VoIP in a LTE network, the first basic evaluation can be done in terms of maximum E2E delay.

2. Simulation Design and Implementation

2.1. Network Traffic Generation

In order to create an application in OPNET, an object is presented which is called Application Definition attribute. This attribute consists of predefined applications that can be customized as per the demands of the user. In Application Definition attribute, there are several predefined applications i.e. HTTP, E-mail, Video, FTP, Voice, Database etc. There are two applications (FTP and VoIP) that are defined in the simulation by using the applications attributes. FTP application is modeled as background traffic in our network model. The FTP configuration parameter, as defined in Table 1 is used during the configuration of FTP. On the other hand, in terms VoIP traffic configuration, the VoIP application uses G.711 encoder scheme where ToS (Type of Service) is set to Interactive Voice (6). After configuring these two applications, it is necessary to configure the Profile Definition attribute in which the behavior of the application is set up. According to the Profile Definition configuration, voice traffic starts at 100 seconds (i.e., offset 60 seconds + start time 40 seconds) and the VoIP application is repeated continuously till the end of the simulation. The profile Definition for VoIP traffic is configured in a way that every VoIP call is added after a fixed time interval. The first VoIP call is established at 100 s (second) of the simulation, after that for each 1 second, one VoIP call is added. The addition of VoIP calls are prepared by repeating the voice application for every 1 second in the Profile Definition. This procedure continues till the end of simulation. In this approach the VoIP calls are increased continuously at fixed interval to the network model.

Table 1 FTP traffic configuration parameters

Attribute	Value
Command Mix	50%
Inter Request Time (Seconds)	Constant(60)
File Size (Bytes)	Constant(1000000000)
Symbolic Server Name	FTP Server
Type of Service	Best Effort(0)
RSVP Parameters	None
Back-End Custom Application	Not Used

2.2. Simulation General Parameters

Table 2 demonstrates the LTE general parameters used in the process of all simulation models of the study. One of the other important entities is the mobility configuration, which is used to determine the mobility model of the workstations. The random waypoint model is a random-based mobility model which has been used in our simulation. There are several appropriate parameters such as speed start time, stop time and pause time which properly control the movement of the workstations. In the simulation, the speed of start time and pause time are set as constant (10 s) and constant (100 s), respectively. On the other hand, the speed of stop time is defined at the end of simulation.

Table 2 LTE Configuration Parameters

LTE Parameter	Value	LTE Parameter	Value
QoS Class Identifier (Voice)	1(GBR)	UL Base Frequency (GHz)	1920 MHz
QoS Class Identifier (FTP)	6(Non-GBR)	UL Bandwidth (MHz)	20 MHz
Uplink Guaranteed Bit Rate (bps)	1 Mbps	UL Cyclic Prefix Type	7 symbols per slot
Downlink Guaranteed Bit Rate (bps)	1 Mbps	DL Base Frequency (GHz)	2110 MHz
Uplink Maximum Bit Rate (bps)	1 Mbps	DL Bandwidth (MHz)	20 MHz
Downlink Maximum Bit Rate (bps)	1 Mbps	DL Cyclic Prefix Type	7 symbols per slot

2.3. Network Model Design

3 network scenarios have been prototyped followed by the example network topology depicted in Figure 1. In all three network scenarios, the area of one cell radius is kept as 1 km. The network elements used during the design of the simulation models include LTE configuration, application configuration, profile configuration and mobility configuration. The other network elements are eNodeB, EPC (Evolved Packet Core) and Workstations.

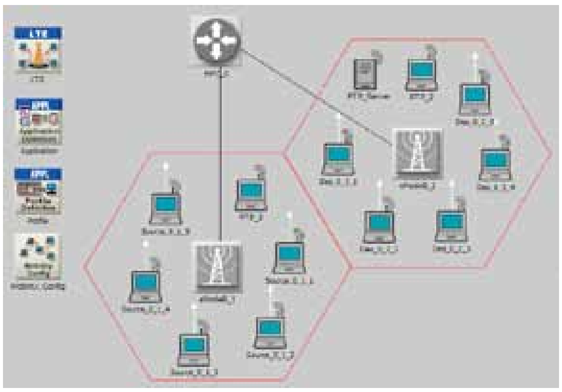


Figure 1 Example Network Topology

2.3.1. Scenario 1: Baseline_VoIP_Network (Low Network Load)

In scenario 1, two eNodeB namely eNodeB_1 and eNodeB_2 are connected with EPC using PPP_DS3 links operated at 44.736 Mbps. Each eNodeB have five VoIP workstations where the nodes under eNodeB_1 serve as the source nodes while the nodes under eNodeB_2 serve as destination nodes. 50% traffic is generated by adding the VoIP calls in way that first call starts at 100 s (second) of the simulation then in every 1 s, a VoIP call is added until the simulation is ended. There are four simulation cases based on the movement of the nodes which are listed in Table 6 (see Appendix A).

2.3.2. Scenario 2: VoIP_Network with 95% network load

About 95% traffic is generated in this scenario which is only voice traffic. The main objective of this scenario is to investigate the effect on voice when the traffic load is about 95%. Table 6 (see Appendix A) represents the simulation cases for scenario 2 that are based on the different mobility speeds of the workstations.

2.3.3. Scenario 3: Congested_VoIP_ with_ FTP_Network (High Network Load)

In scenario 3, mixed traffic (i.e., VoIP and FTP) is generated where FTP traffic is modeled in order to introduce background traffic in the simulation. Guaranteed Bit Rate (GBR) is assigned to the voice traffic while Non-Guaranteed Bit Rate (Non-GBR) is assigned to the FTP traffic. One FTP workstation is created in each cell. The FTP application parameters are shown in previous section. It is important to mention that FTP workstations are fixed in all the simulation cases. Table 6 depicted in Appendix A shows the simulation cases considering two types of criteria i.e. the mobility of the workstations and various background traffic loads (FTP).

3. Results and Analysis

3.1. E2E Delay Performance

E2E delay refers to the time needed for a packet to be transmitted from one node to another node in a network. Generally in a VoIP network, three types of delay occur when the packet traverses through the network. Firstly, sender's delay, when packets are traversed from source node. The other two are network delay and receiver delay. For VoIP applications, the packet E2E delay should not exceed 150 ms (millisecond) to evaluate that the quality of the created VoIP calls are accepted [9]. In all the scenarios, the sources and destinations happen to be started at 100 s as the start time of the profile and application configuration have been set to 40 s and 60 s, respectively. In all of the figures presented in following sections, X axis represents simulation time in min (minute) while Y axis represents E2E delay in second.

3.2. E2E Delay Performance for Baseline_VoIP_ Network (Scenario 1)

In Figure 2, the blue line shows the E2E delay at the fixed node speed of 0 mps (meter per second) while red, green and turquoise lines show the E2E delay at the node speeds of 10 mps, 20 mps and 50 mps , respectively. As it can be seen from Figure 2 and observed in Table 3, case 1 has the lowest E2E delay of about 73 ms on an average varying from 73.38 ms to 76.06 ms. On the other hand, the E2E delay for case 4 is about 73.82 ms on an average, attaining a maximum delay among all the cases.

However, at the initial stage of the simulation time, the E2E delay for case 2 is found to approximately be 76.06 ms, which is even higher than case 4. With the increasing simulation time, it settles around 73.38 ms, and remains there for the rest of the time.

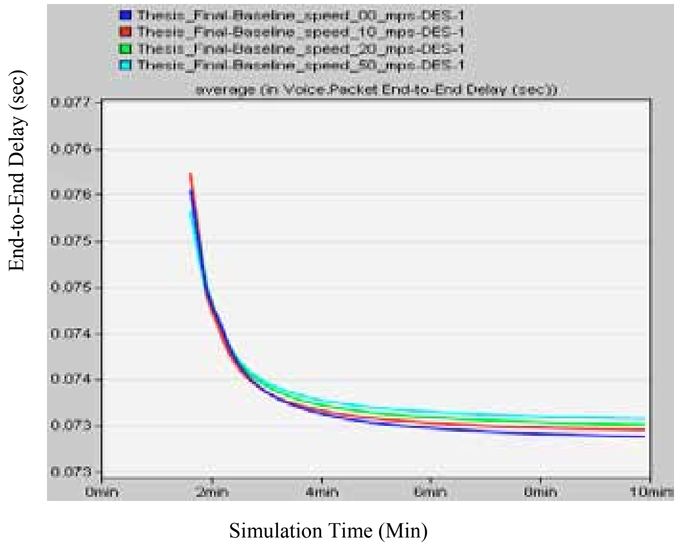


Figure 2 E2E delay for scenario 1

In addition, the average E2E delay for case 4 is approximately 0.25% higher than that of case 1 scenario. All VoIP calls are accepted due to the packet E2E delay doesn't exceed the threshold value of 150 ms.

Table 3 Summary statistics of E2E delay of scenario 1

Scenario	**Min. [s]**	**Avg. [s]**	**Max. [s]**	**Std Dev [s]**
Case 1	7.338E-02	7.368E-02	7.606E-02	5.10E-04
Case 2	7.345E-02	7.372E-02	7.622E-02	4.97E-04
Case 3	7.351E-02	7.378E-02	7.606E-02	4.85E-02
Case 4	7.357E-02	7.382E-02	7.582E-02	4.26E-02

E2E Delay performance for VoIP_Network with Medium Network Load (Scenario 2)

Figure 3 shows a graphical representation of a comparative analysis on the E2E delay under 4 mobility cases of scenario 2. As mentioned earlier, there are four different scenarios based on the movement of the node where case 1 is designed with fixed node while case 2, 3 and 4 are designed with node speed at 10, 20 and 50 mps, respectively. The comparison against different cases scenarios in terms of E2E delay is observed in Figure 3 and referred to Table 4.

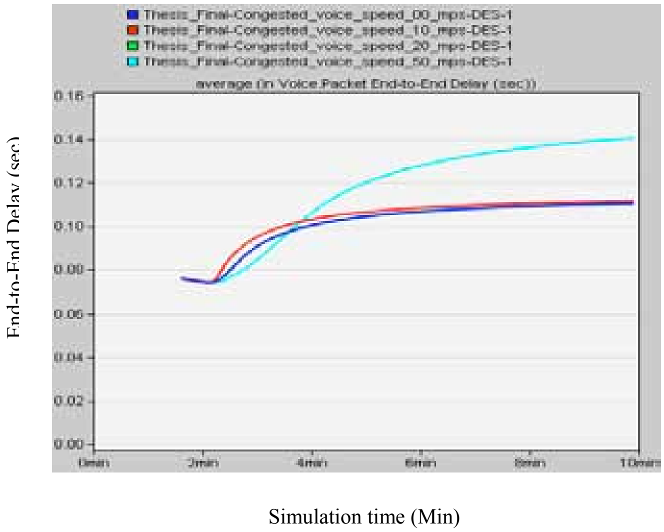


Figure 3 E2E delay performance for scenario 2

. In the Figure 3, the blue line shows the E2E delay of the case 1 scenario while red, green and turquoise lines show the E2E delay of case 2, case 3 and case 4 scenarios, respectively. As can be seen in Figure 3, the VoIP network with 95% network load, with a fixed node speed, maintain a lower E2E delay level of about 101.283 ms on average and varies from 74 ms to 110 ms. The topmost curve represents the case 4 scenarios, attaining a maximum delay among all the case scenarios. The E2E delay for this case is about 116 ms on average. Case 4 E2E delay is lower than other cases between simulation times 135 s to 230 s. The E2E delays are overlapped in case 2 and case 3. Although at the initial stage of the simulation time, the E2E delay is higher than case 4. With the increasing simulation time, the both curves settle at 111 ms.

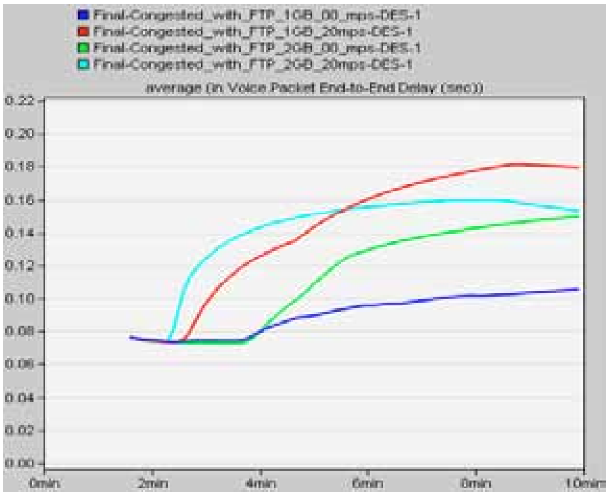
In scenario 2, the average delay of case 4 is approximately 13% higher than case 1. Meanwhile, the average E2E delay of case 2 and 3 is approximately 11 % lower than that of case 4. In all scenarios, all VoIP calls are accepted due to the packet E2E delay doesn't exceed the threshold value of 150 ms.

Table 4 Summary statistics of E2E delay of Congested VoIP Network

Scenario	Min. [s]	Avg. [s]	Max. [s]	Std Dev [s]
Case 1	7.423E-02	1.012E-01	1.105E-01	1.128E-02
Case 2	7.423E-02	1.032E-01	1.116E-01	1.103E-02
Case 3	7.423E-02	1.032E-01	1.116E-01	1.103E-02
Case 4	7.429E-02	1.167E-01	1.405E-01	2.287E-02

3.3. E2E Delay performance for Congested_VoIP_ with_ FTP_Network (Scenario 3)

Figure 4 illustrates the comparable performance of the E2E delay under the 4 different mobility cases against highly congested network scenario.



End-to-End Delay (sec)

Simulation Time (Min)

Figure 4 E2E delay performance for scenario 3

In this scenario, our observation is to determine the effect of voice data when the FTP traffic is used to increase the load of different scenarios. The comparison against different case scenarios in terms of E2E delay is referred to Table 5 and observed in Figure 4.

In Figure 4, the blue line shows the E2E delay of the case 1, while red, green and turquoise lines show the E2E delay of case 2, case 3 and case 4, respectively. For VoIP workstations, the speed of case 1 and case 3 are fixed (0 mps), while the mobility rate is 20 mps in case 2 and case 4.

In case 1, the E2E delay varies from 73 ms to 105 ms attaining an average value of 90 ms. the average E2E delay in such a scenario is about 143 ms, which is about 36% higher than case 1 scenario. For the case 2 scenario, the E2E delay is appeared to be higher than that of case 1 scenario. From Table 5, it can be seen that the minimum E2E delay value in case 2 is 73 ms while the maximum value is found to be 181 ms. For the case 3, a lower E2E delay is maintained between 100 s to 240 s, after that the curve rises gradually till end of the simulation. It can be observed that the minimum E2E delay is about 72 ms, while, the maximum E2E delay is found to be approximately 149 ms. The average E2E delay is about 113 ms in this scenario. In the case 4, the E2E delay varies from 73 ms to 159 ms and the E2E delay for this case is 141 ms in average. In the "VoIP congested with FTP Network", the average delay of case 4 is approximately 19% higher than that of case 3.

Table 5 Summary statistics of E2E delay of VoIP Congested with FTP Network

Scenario	Min. [s]	Avg. [s]	Max. [s]	Std Dev [s]
Case 1	7.367E-02	9.067E-02	1.054E-01	1.161E-02
Case 2	7.331E-02	1.433E-01	1.812E-01	3.737E-02
Case 3	7.265E-02	1.139E-01	1.495E-01	3.031E-02
Case 4	7.341E-02	1.413E-01	1.595E-01	2.642E-02

3.4. Conclusion

In this research work, an effective study, analysis and evaluation of the E2E delay performance evaluation for VoIP traffic under varying mobility speed in the LTE network was carried out. The evaluation was made by simulating three network scenarios i.e., Baseline VoIP network with 50% network load, VoIP network with 95% network load and VoIP congested with FTP network scenarios. Four cases under each scenario were evaluated, one case with stationary node and other three cases with mobile nodes (gradually increasing the node speed). The simulation results for baseline VoIP network (non-congested), congested VoIP network and congested VoIP with FTP network show that as the speed of node is gradually increased, E2E delay is slightly increased.

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Appendix A.

A.1. Mobility and Network Load Configuration Parameters for Scenarios 1,2, and 3

There are four simulation cases based on the movement of the nodes for each scenario listed in Table 6.

Table 6 Simulation case definition parameters of Scenarios 1, 2, and 3

Simulation case definition of Baseline VoIP Network (Scenario 1)					
Case	**Bandwidth (MHz)**	**VoIP Traffic Load (%)**	**Cell Radius (Km)**	**FTP File Size (Bytes)**	**Speed (m/s)**
1	20	50	1		Fixed(0)
2	20	50	1		10
3	20	50	1		20
4	20	50	1		50
Simulation case definition of Congested VoIP Network (Scenario 2)					
1	20	95	1		Fixed (0)
2	20	95	1		10
3	20	95	1		20
4	20	95	1		50
Simulation case definition of VoIP Congested with FTP Network (Scenario 3)					
1	20	80		Constant (10,00000000)	Fixed
2	20	80		Constant (10,00000000)	20
3	20	80		Constant (20,00000000)	Fixed
4	20	80		Constant (20,00000000)	20